(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 622 779 B1

(12) # EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
12.06.2002 Bulletin 2002/24

(51) Int Cl.7: **G10K 11/16**, G10K 11/178

(21) Application number: **94106495.8**

(22) Date of filing: **26.04.1994**

(54) **Multiple adaptive filter active noise canceller**

Aktiver Lärmdämpfer mit vielfachadaptivem Filter

Dispositif d'atténuation actif du bruit muni d'un filtre adaptif multiple

(84) Designated Contracting States:
DE FR GB IT

(30) Priority: **27.04.1993 US 53728**

(43) Date of publication of application:
02.11.1994 Bulletin 1994/44

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Printed by Jouve, 75001 PARIS (FR)

Description

BACKGROUND OF THE INVENTION

[0001] The present invention as it is defined in the appended claims relates to active noise cancellation systems, and more particularly to systems having extended frequency stability regions so as to permit the suppression of broader bandwidth disturbances.

[0002] The objective in active noise cancellation is to generate a waveform that inverts a nuisance noise source and suppresses it at selected points in space. In active noise cancelling, a waveform is generated for subtraction, and the subtraction is performed acoustically, rather than electrically.

[0003] In a basic active noise cancellation system, a noise source is measured with a local sensor such as an accelerometer or microphone. The noise propagates acoustically over an acoustic channel to a point in space where noise suppression is desired, and at which is placed another microphone. The objective is to remove the acoustic energy components due to the noise source. The measured noise waveform from the local sensor is input to an adaptive filter, the output of which drives a speaker. The second microphone output at the point to be quieted serves as the error waveform for updating the adaptive filter. The adaptive filter changes its weights as it iterates in time to produce a speaker output that at the microphone looks as much as possible (in the minimum mean squared error sense) like the inverse of the noise at that point in space. Thus, in driving the error waveform to have minimum power, the adaptive filter removes the noise by driving the speaker to generate inverted noise in order to suppress it.

[0004] Many previous active noise cancelers use the filtered-X LMS algorithm, which requires a training mode. The function of the training mode is to learn the transfer functions of the speaker and microphones used in the system so that compensation filters can be inserted in the feedback loop of the LMS algorithm to keep it stable. As the physical situation changes, the training mode must be reinitiated. For example, in an automobile application to suppress noise within a passenger compartment, the training mode may need to be performed again every time a window is opened, or another passenger enters the compartment, or when the automobile heats up during the day. The training mode can be quite objectionable to passengers in the vehicle.

[0005] U.S. Patent 5,117,401 describes an active adaptive noise canceller which does not require a training mode. The insertion of a time delay in the computation of the updated weights modifies the frequency stability regions of the canceller. Hence, the canceller provides a mechanism through which the adaptive noise cancellation can be easily adapted to suit any application at hand by simply adjusting the time delay value to acquire the desired frequency stability regions. This approach however, has a limitation in that the insertion of delay provides very limited control over the bandwidth of the frequency stability region.

[0006] GB 2 257 327 A discloses an active vibration control system for suppressing vibrations or noise. In order to provide satisfactory noise suppression effects over the entire frequency range, the system comprises a plurality of channels for different frequency bands. Each channel comprises a bandpass filter and an adaptive control circuit. The output signals of all channels are added to generate an error signal ε. The error signal is used to vary the inverse transfer characteristic of the channels.

[0007] It is the object of the invention to provide an active noise canceller system having an increased noise cancellation bandwidth.

[0008] This object is solved by the invention as claimed in independent claim 1.

[0009] Preferred embodiments are defined by the dependent claims.

[0010] The invention provides an active noise cancellation system employing a LMS filter algorithm with extended frequency stability regions to permit the suppression of broader bandwidth disturbances.

[0011] In accordance with the invention, an active noise canceller is described, wherein the noise bandwidth over which suppression is to take place is partitioned into frequency sub-bands, and multiple adaptive filter channels using different delays to achieve stability in the respective sub-bands are employed. Each channel includes bandpass filters to restrict the channel to operation over only the particular frequency sub-band, and delay is inserted in the operation of the filter weight updating. Because each channel is stable over its frequency sub-band, the canceller operates over the extended noise bandwidth formed by all the sub-bands.

[0012] In an exemplary embodiment, the canceller suppresses noise signals from a noise source, and includes a noise sensor for generating noise sensor signals representative of the noise signals, an acoustic sensor, and acoustic output device. First and second channels are responsive to the noise sensor signals and the acoustic sensor signals, and adaptive filters generate respective channel output signals which are combined to drive the acoustic output device. Each channel includes respective bandpass filters which restrict the operation of the channel to a particular frequency subband, by filtering the noise sensor signal and the acoustic sensor signal. Each channel further includes delay means for delaying the operation of the filter weight updating.

BRIEF DESCRIPTION OF THE DRAWING

[0013] These and other features and advantages of the present invention will become more apparent from the following detailed description of an exemplary embodiment thereof, as illustrated in the accompanying drawings, in which:

[0014] FIG. 1 illustrates, in the frequency domain, an adaptive noise canceller (ANC) employing a delay in the weight updating to remove the necessity for a training mode.

[0015] FIG. 2 illustrates, for the canceller of FIG. 1, the phase response of the product of the speaker-microphone and time delay transfer functions.

[0016] FIG. 3 is a simplified schematic block diagram of an adaptive noise cancellation system with parallel ANC processing channels to extend the frequency stability regions.

[0017] FIG. 4 is a simplified schematic block diagram of an ANC processing channel comprising the system of FIG. 3.

[0018] FIGS. 5-7 illustrate ANC systems for reducing electrical motor/engine noise, reducing engine noise and enhancing audio program deliveries, respectively, in accordance with the invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

[0019] FIG. 1 depicts the frequency domain analog, for explanatory purposes, of an adaptive noise canceller (ANC) 50, more fully described in U.S. Patent 5,117,401, which does not require a training mode. The frequency domain analog is discussed to illustrate the frequency stability regions of this canceller. The noise x (n) from a noise source is passed through a fast Fourier transform (FFT) function, and the resulting FFT components $x_\omega(n)$ are passed through the acoustic channel, represented as block 54, with a channel transfer function $P(j\omega)$. The ANC system 50 includes a microphone 58 with its transfer function $H_M(j\omega)$ and a speaker 60 with its transfer function $H_s(j\omega)$. The acoustic channel 54 inherently performs the combining function 56 of adding the channel response to the negative of the speaker excitation. The microphone 58 responds to the combined signal from combiner 56. The Fourier components are also passed through an adaptive LMS filter 62 with transfer function $G(j\omega)$. The filter weights are updated by the microphone responses, delayed by a time delay Δ (66).

[0020] It can be shown that the adaptive filter 62 of the ANC system 50 of FIG. 1 is stable in the frequency regions in which the real part of the product of the microphone-speaker and the delay line transfer functions is positive, i.e., $\mathrm{Real}\{\exp(j\omega\Delta)H_m(j\omega)H_s(j\omega)\}>0$. A corollary to this inequality is that the phase of $\{\exp(j\omega\Delta))H_m(j\omega)H_s(j\omega)\}$ must lie inside $(2n\pi-\pi/2, 2n\pi+\pi/2)$, n=1, 2, ..., i.e., the right side of the complex plane. The phase of $\{\exp(j\omega\Delta)H_m(j\omega)\ H_s(j\omega)\}$ is plotted in FIG. 2, where $H_m(j\omega)$ and $H_s(j\omega)$ are modelled by a Tchebychev and a Butterworth filter, respectively. In this example for the "no delay" case, i.e., Δ=0, the stability regions of the adaptive filter can be found by locating the phase of $\{\exp(j\omega\Delta)H_m(j\omega)H_s(j\omega)\}$ within the stippled bands of FIG. 2, and they fall approximately from 1 to 2 Hz, 17 to 42 Hz, 70 to 170 Hz, 1500 to 2900 Hz, and 3400 to 5000 Hz. For a sampling frequency of 10,000 Hz, the insertion of a 7 sample delay provides upward bending of the phase curve to the speaker-microphone phase response function so that the stability regions now have changed to approximately 1 to 2 Hz, 17 to 42 Hz, 70 to 1400 Hz and 3000 to 5000 Hz.

[0021] "Frequency stability region" in the context of this ANC system means that the adaptive filter is stable when operated to suppress disturbing signals within this frequency range. Conversely, the adaptive filter cannot be kept stable absolutely when it is excited by signals that fall outside of this region.

[0022] In the example shown in FIG. 2, the insertion of a 7 sample delay, based on a sampling frequency of 10,000 Hz, has extended the frequency stability region to from 70 to 1400 Hz, as compared to the region 70 to 170 Hz with no delay. However, further expansion of the frequency stability region beyond the 1400 Hz is not achievable with the use of a single insertion of delay. This is because a bulk delay has a phase response of a straight line with its slope proportional to the delay value. Consequently, there is a limited range of frequencies for which a single value of the bulk delay can stabilize the composite phase response of the system. On the other hand, if the disturbance signal is partitioned, in accordance with this invention, into two (or more) separate frequency bands prior to input to two adaptive filters which are structured to operate independently in parallel with two different delays, it is then possible to suppress a disturbing signal which has frequency components higher than 1400 Hz.

[0023] FIG. 3 depicts a block diagram of an ANC system 100 implemented in the time domain and embodying this multiple adaptive filter scheme. ANC system 100 operates to cancel noise acoustic energy generated by a noise source 90, which propagates over an acoustic channel indicated by block 92, by generating acoustic cancelling energy with a speaker 152. The acoustic channel inherently subtracts the acoustic energy emitted by ANC speaker 152 from the noise energy emitted by source 90. The system 100 includes a microphone 154 which detects the error, i.e., the residual acoustic energy, and feeds back an electrical error signal to the ANC signal processing channels 120 and 140. The system 100 further includes a sensor 110 for sensing the noise energy emitted by the source 90. The sensor output signal is fed to the channels 120 and 140 which operate over different portions of the frequency band. The outputs of the respective channels 120 and 140 are summed at node 150 to cancel over a larger bandwidth than either channel could separately, and the combined output drives the speaker 152.

[0024] The ANC system 100 of FIG. 3 effectively partitions the disturbance signal band into two separate frequency bands, with one adaptive filter operating in one band, and the other adaptive filter operating in the second band. This partition is achieved with the use of two

pairs of matching bandpass filters at the inputs to the adaptive filters and the output of the error microphone. These pairs of bandpass filters should have pass band characteristics that are consistent with their respective frequency stability regions so that the adaptive filters are not excited by out-of-band energy thereby resulting in filter instability.

[0025] FIG. 4 illustrates the ANC signal processing channel 120 in further detail. Channel 140 is similar to channel 120, except that the bandpass filters are tuned to a different frequency band, and accordingly need not be described further in detail. Channel 120 includes a pair of bandpass filters 121 and 130. Filter 121 filters the signal from the noise source sensor 110, and filter 130 filters the signal from the error microphone 154. The filters are constructed to have identical pass bands. The filtered signals are digitized by respective A/D convertors 122 and 131. The digitized signal from convertor 122 drives a recursive adaptive LMS filter 138 which employs the LMS algorithm. The filter 138 comprises a feed-forward adaptive filter 123, a feed-backward adaptive filter 132, and a summing node 124, and is updated in the manner described in "An Adaptive Recursive LMS Filter," by P.L. Feintuch, IEEE Proceedings, Vol. 64, No. 11, November 1976. The signal from convertor 122 is also delayed by delay 125, and the delayed digitized signal is an input to the weight update logic 126. The digitized signal from convertor 131 is provided as an input to the weight update logic 126 and to the weight update logic 134.

[0026] The weight update logic 123 serves to provide the updated weights for the adaptive LMS filter 123. The filter 123 output is summed at summing node 124 with the output from adaptive filter 132 in a recursive relationship, with the summed signal driving the filter 132. The summed signal also is delayed by delay 133, and then provided to the weight update logic 134 as another input. The digital summed signal is also converted into an analog signal by digital-to-analog convertor (DAC) 135. The converted analog signal is in turn summed with the outputs from the other channel 140 at combiner 150, and the combined signal from both channels drives the cancelling speaker 152.

[0027] The channel 120 operates in the same manner as the recursive noise canceller system 40 shown in FIG. 4 of U.S. Patent 5,117,401, except that the system 40 does not employ bandpass filters as in channel 120.

[0028] For the exemplary embodiment in FIGS. 3 and 4, consider the case where the bandwidth of the disturbance is from 70 to 3200 Hz. An ANC system comprising one adaptive filter will not be capable of handling the bandwidth since there is no single delay value that can provide sufficient phase compensation over a bandwidth of that size. Using the invention described herein, it is now possible to do so. For this example, bandpass filters 121 and 130 have bandwidth of 70 to 1300 Hz. The corresponding bandpass filters for channel 140 have a bandwidth of 1300 to 3200 Hz. Delay circuits 125 and 133 introduce a delay equal to 7 samples (at a sample rate of 10,000 Hz), while the corresponding delay circuits for channel 140 introduces a delay equivalent to 4 samples (see FIG. 2 for the phase response of these delay values). This will provide active noise suppression over the entire 70 to 3200 Hz band without requiring a training mode. This invention can be further generalized to have a structure which contains multiple parallel adaptive filters.

[0029] FIG. 5 illustrates a first exemplary application for an ANC system 200 in accordance with the invention. In this application the system 200 is used to cancel noise from a noise source such as an electric motor or an engine 190. Here, a reference sensor 202 is used to measure the noise signals from the noise source 190. The error microphone 204 is placed at the point in space at which the noise signal is to be cancelled. A speaker 206 is placed adjacent the noise source 190, and is connected to the ANC signal processing circuit 210 which drives the speaker with appropriate drive signals so as to produce cancelling signals which cancel the noise from the noise source 190.

[0030] The ANC circuit 210 comprises the first and second ANC channels 120 and 140 and adder 150 of the system shown in FIG. 3. Circuit 210 receives input signals from the reference sensor 202 and the error microphone 204.

[0031] FIG. 6 shows a second exemplary application for an ANC system 250 in accordance with the invention, used to reduce the engine noise emitted from an automobile engine 240 via the automobile tailpipe 245. In this system, the reference sensor 252 is placed adjacent the engine, and the error microphone is place adjacent the tailpipe 245 near the tailpipe opening. The speaker 256 is located in an opening in the tailpipe between the engine and the error microphone 254, for emitting an anti-noise soundwave to cancel engine noise. The speaker 256 is driven by the ANC signal processing circuit 260. The circuit 260 receives input signals from the reference sensor 252 and the error microphone 254. The ANC circuit 260 comprises the first and second ANC channels 120 and 140 and adder 150 of the system of FIG. 3.

[0032] FIG. 7 shows a third exemplary application for an ANC system 300 in accordance with the invention, used in a stereo headphone set 290 to cancel a disturbing noise soundwave. In this system, the headphone speakers 306 are used to produce the reduced noise soundwave. A reference microphone 302 is attached to the headphone bridge element connecting the respective ear pieces. The error microphones 304A and 304B are attached adjacent the respective speakers 306A and 306B to sense the reduced noise soundwave. In this system, the outputs from the respective ANC signal processing circuits 308A and 308B are added by adders 300A and 300B to the respective left and right audio data signals, provided as a communication message or music from left and right sources 295A and 295B. The com-

bined signal in the respective channel drives the respective headphone speaker 306A and 306B. Each ANC signal processing circuit 308A and 308B, as in the preceding examples, comprises ANC channels 120 and 140 and adder 150 of FIG. 3. The circuits 308A and 308B receives input signals from the respective reference sensor 302A or 302B and the error microphone 304A or 304B. The ANC circuits generate a noise cancelling waveform which drives a respective speaker 306A or 306B, along with the desired sound waveform from the respective source 295A or 295B. Of course, the invention may be used with a monaural headphone set, requiring only a single ANC signal processing channel.

[0033] It is understood that the above-described embodiments are merely illustrative of the possible specific embodiments which may represent principles of the present invention. For example, a noise canceller in accordance with the invention can alternatively be implemented in the frequency domain. Other arrangements may readily be devised in accordance with these principles by those skilled in the art without departing from the scope of the invention as defined in the appended claims.

Claims

1. An active noise canceller system (100) for suppressing noise over a predetermined noise bandwidth, comprising:

   a noise sensor (110) for generating a noise sensor signal indicative of said noise to be suppressed,

   an error sensor (154) for generating an error signal,

   an acoustic output device (152) for generating a canceling acoustic signal,

   a plurality of adaptive filter channels (120, 140) responsive to said noise sensor signal and said error signal, each channel restricted to operation over a predetermined frequency sub-band comprising said noise bandwidth and producing a channel output signal; and

   means (150) for combining said plurality of channel output signals to provide a combined signal for driving said acoustic output device (152) to generate said canceling acoustic signal,

   characterized in that
   each channel employs delay in the updating of adaptive filter weights to achieve stability in operation in said frequency sub-band over which said channel operates,
   wherein the respective delay values for the respective channels are different delay values.

2. A canceller system according to Claim 1, further **characterized in that** each channel (120, 140) further comprises bandpass filter means (121, 130) for filtering said noise sensor signal and said error signal so as to pass only signal frequency components within respective frequency sub-band for said channel, thereby restricting said channel to operation over said frequency sub-band.

3. A canceller system according to any preceding claim, further **characterized in that** each said channel (120, 140) comprises recursive adaptive filter means (138).

4. A canceller system according to any preceding claim, further **characterized in that** said frequency sub-bands cover said noise bandwidth.

5. A canceller system according to any preceding claim, further **characterized in that** each channel (120, 140) further comprises delay means (125) for providing a delayed version of said noise sensor signal delayed by a predetermined delay, adaptive filter weight update logic means (126) responsive to said delayed version of said noise sensor signal for updating adaptive filter weight inputs to adaptive filter means (123) comprising said channel.

6. A canceller system according to Claim 1, wherein said plurality of adaptive filter channels is further **characterized by**:

   a first cancellation channel (120) coupled to said noise sensor (110) and said acoustic sensor (154), said first channel comprising a first bandpass filter means (121) for filtering said noise sensor signals, said first filter having a first pass band, a second bandpass filter means (130) for filtering signals generated by said acoustic sensor, said second filter having said first pass band, a first delay means (125) for delaying said first bandpass filtered noise sensor signals by a preselected first time delay, and first adaptive filter means having a plurality of inputs coupled to said first and second bandpass filter means (121, 130) and said first delay means (125), and providing a first filter output; and
   a second cancellation channel coupled to said noise sensor and said acoustic sensor, said second channel comprising a third bandpass filter means for filtering said noise sensor signals, said third filter having a second pass band, fourth bandpass filter means for filtering

said acoustic sensor signals, said fourth filter having said second pass band, second delay means for delaying said third bandpass filtered noise sensor signals by a preselected second time delay, and second adaptive filter means having a plurality of inputs coupled to said second bandpass filter means, said acoustic sensor and said second delay means, and providing a second filter output.

7. A canceller according to Claim 6 wherein said first adaptive filter means comprises a plurality of first filter weights, and first weight update logic means (126) responsive to said second bandpass filtered signals from said acoustic sensor for adjusting said first filter weights, said second adaptive filter means comprises a plurality of second filter weights, and second weight update logic means responsive to said fourth bandpass filtered signals from said acoustic sensor for adjusting said second filter weights.

8. A canceller according to Claim 7, further **characterized in that** said first time delay does not equal said second time delay.

9. A canceller system according to Claim 7 or Claim 8, further **characterized in that** said first and second filter output signals are digitized signals, and said combining means (150) comprises a digital adder means.

10. A canceller system according to Claims 7, 8 or 9, further **characterized in that** said first adaptive filter means comprises a recursive adaptive filter means (138) comprising:

a first adaptive filter (123) responsive to said first bandpass filtered noise sensor signals and comprising a plurality of first adaptive filter weight inputs, said first adaptive filter providing a first adaptive filter output;
a first weight update logic means (126) responsive to said delayed first bandpass filtered noise sensor signals and to said second bandpass filtered acoustic sensor signals for adaptively updating said first adaptive filter weight inputs;
a second adaptive filter (132) for providing a second adaptive filter output;
means (124) for combining said first and second adaptive filter outputs to provide said first filter output;
said second adaptive filter responsive to said first filter output and comprising a plurality of second adaptive filter weight inputs;
third delay means (133) for providing a delayed version of said first filter output which is delayed by a third predetermined time delay;
a second weight update logic means (134) responsive to said delayed version of said first filter output and to said second bandpass filtered acoustic sensor signals for adaptively updating said second adaptive filter weight inputs.

Patentansprüche

1. Aktives Schallunterdrückungssystem (100) zum Unterdrücken von Schall über eine vorbestimmte Schallbandbreite, umfassend:

einen Schallsensor (110) zum Erzeugen eines Schallsensorsignales, das den zu unterdrükkenden Schall angibt,

einen Fehlersensor (154) zum Erzeugen eines Fehlersignals,

ein akustisches Ausgabegerät (152) zum Erzeugen eines unterdrückenden akustischen Signales,

eine Vielzahl von Adaptivfilterkanälen (120, 140), die auf das Schallsensorsignal und das Fehlersignal ansprechen, wobei jeder Kanal auf einen Betrieb über ein vorbestimmtes Frequenzunterband, das die Schallbandbreite umfasst, eingeschränkt ist, und ein Kanalausgabesignal erzeugt; und

Mittel (150) zum Kombinieren der Vielzahl von Kanalausgabesignalen, um ein kombiniertes Signal zum Treiben des akustischen Ausgabegerätes (152) bereit zu stellen, um das unterdrückende akustische Signal zu erzeugen,

dadurch gekennzeichnet, dass
jeder Kanal bei der Aktualisierung von Adaptivfiltergewichtungen eine Verzögerung anwendet, um Stabilität im Betrieb in dem Frequenzunterband, in dem der Kanal arbeitet, zu erzielen,
wobei die jeweiligen Verzögerungswerte für die jeweiligen Kanäle verschiedene Verzögerungswerte sind.

2. Unterdrückungssystem nach Anspruch 1, weiterhin **dadurch gekennzeichnet, dass**
jeder Kanal (120, 140) ferner Bandpassfiltermittel (121, 130) umfasst zum Filtern des Schallsensorsignals und des Fehlersignals, um nur Signalfrequenzkomponenten innerhalb des jeweiligen Frequenzunterbandes für den Kanal durchzulassen, wodurch der Kanal auf dem Betrieb in dem Frequenzunterband eingeschränkt wird.

3. Unterdrückungssystem nach einem der vorhergehenden Ansprüche, weiterhin
dadurch gekennzeichnet, dass
jeder Kanal (120, 140) rekursive Adaptivfiltermittel (138) umfasst.

4. Unterdrückungssystem nach einem der vorhergehenden Ansprüche, weiterhin
dadurch gekennzeichnet, dass
die Frequenzunterbänder die Schallbandbreite abdecken.

5. Unterdrückungssystem nach einem der vorhergehenden Ansprüche, weiterhin
dadurch gekennzeichnet, dass
jeder Kanal (120, 140) ferner Verzögerungsmittel (125) umfasst zum Bereitstellen einer verzögerten Version des Schallsensorsignals, das um eine vorbestimmte Verzögerung verzögert ist, und Adaptivfiltergewichtungsaktualisierungslogikmittel (126), die auf die verzögerte Version des Schallsensorsignals ansprechen, zum Aktualisieren der Adaptivfiltergewichtungseingabesignale für die Adaptivfiltermittel (123), die den Kanal umfassen.

6. Unterdrückungssystem nach Anspruch 1, wobei die Vielzahl von Adaptivfilterkanälen weiterhin
gekennzeichnet ist durch:

einen ersten Unterdrückungskanal (120), der mit dem Schallsensor (110) und dem akustischen Sensor (154) verbunden ist, wobei der erste Kanal erste Bandpassfiltermittel (121) zum Filtern der Schallsensorsignale umfasst, wobei der erste Filter ein erstes Durchlassband aufweist, wobei der erste Kanal ferner zweite Bandpassfiltermittel (130) umfasst zum Filtern von Signalen, die von dem akustischen Sensor erzeugt sind, wobei der zweite Filter das erste Durchlassband aufweist, wobei der erste Kanal ferner erste Verzögerungsmittel (125) umfasst zum Verzögern der ersten bandpassgefilterten Schallsensorsignale um eine vorausgewählte erste Zeitverzögerung, und wobei der erste Kanal weiterhin erste Adaptivfiltermittel umfasst, die eine Vielzahl von Eingängen aufweisen, die mit den ersten und zweiten Bandpassfiltermitteln (121, 130) und den ersten Verzögerungsmitteln (125) verbunden sind, und ein erstes Filterausgabesignal liefern; und

einen zweiten Unterdrückungskanal, der mit dem Schallsensor und dem akustischen Sensor verbunden ist, wobei der zweite Kanal dritte Bandpassfiltermittel umfasst zum Filtern der Schallsensorsignale, wobei der dritte Filter ein zweites Durchlassband aufweist, wobei der zweite Kanal weiterhin vierte Bandpassfiltermittel zum Filtern der akustischen Sensorsignale umfasst, wobei der vierte Filter das zweite Durchlassband aufweist, wobei der zweite Kanal weiterhin zweite Verzögerungsmittel umfasst zum Verzögern der dritten bandpassgefilterten Schallsensorsignale um eine vorausgewählte zweite Zeitverzögerung, und wobei der zweite Kanal weiterhin zweite Adaptivfiltermittel umfasst, die eine Vielzahl von Eingängen aufweisen, die mit den zweiten Bandpassfiltermitteln, dem akustischen Sensor und den zweiten Verzögerungsmitteln verbunden sind, und ein zweites Filterausgabesignal liefern.

7. Unterdrückungssystem nach Anspruch 6, wobei die ersten Adaptivfiltermittel eine Vielzahl von Filtergewichtungen umfassen sowie erste Gewichtungsaktualisierungslogikmittel (126), die auf die zweiten bandpassgefilterten Signale von dem akustischen Sensor ansprechen, zum Einstellen der ersten Filtergewichtungen, wobei die zweiten Adaptivfiltermittel eine Vielzahl von zweiten Filtergewichtungen umfassen, und zweite Gewichtungsaktualisierungslogikmittel, die auf die vierten bandpassgefilterten Signale von dem akustischen Sensor ansprechen zum Einstellen der zweiten Filtergewichtungen.

8. Unterdrückungssystem nach Anspruch 7, weiterhin
dadurch gekennzeichnet, dass
die erste Zeitverzögerung nicht gleich der zweiten Zeitverzögerung ist.

9. Unterdrückungssystem nach Anspruch 7 oder 8, weiterhin
dadurch gekennzeichnet, dass
die ersten und zweiten Filterausgabesignale digitalisierte Signale sind und die Kombiniermittel (150) digitale Additionsmittel umfassen.

10. Unterdrückungssystem nach Anspruch 7, 8 oder 9, weiterhin
dadurch gekennzeichnet, dass
die ersten Adaptivfiltermittel rekursive Adaptivfiltermittel (138) umfassen, die enthalten:

einen ersten Adaptivfilter (123), der auf die ersten bandpassgefilterten Schallsensorsignale anspricht und eine Vielzahl von ersten Adaptivfiltergewichtungseingängen umfasst, wobei der erste Adaptivfilter ein erstes Adaptivfilterausgabesignal bereitstellt;

erste Gewichtungsaktualisierungslogikmittel (126), die auf die verzögerten ersten bandpassgefilterten Schallsensorsignale und die zweiten bandpassgefilterten Akustiksensorsignale ansprechen zum adaptiven Aktualisieren der er-

sten Adaptivfiltergewichtungseingangssignale;

einen zweiten Adaptivfilter (132) zum Bereitstellen eines zweiten Adaptivfilterausgabesignales;

Mittel (124) zum Kombinieren der ersten und zweiten Adaptivfilterausgabesignale, um das erste Filterausgabesignal bereitzustellen;

wobei der zweite Adaptivfilter auf das erste Filterausgabesignal anspricht und eine Vielzahl von zweiten Adaptivfiltergewichtungseingängen aufweist;

dritte Verzögerungsmittel (133) zum Bereitstellen einer verzögerten Version des ersten Filterausgabesignals, das um eine dritte vorbestimmte Zeitverzögerung verzögert ist;

zweite Gewichtungsaktualisierungslogikmittel (134), die auf die verzögerte Version des ersten Filterausgabesignals und auf die zweiten bandpassgefilterten Akustiksensorsignale ansprechen, zur adaptiven Aktualisierung der zweiten Adaptivfiltergewichtungseingabesignale.

Revendications

1. Système d'élimination de bruit actif (100) pour supprimer le bruit sur une largeur de bande de bruit prédéterminée, comprenant :

   un détecteur de bruit (110) pour générer un signal de détecteur de bruit indicatif dudit bruit devant être supprimé,
   un détecteur d'erreur (154) pour générer un signal d'erreur,
   un dispositif de sortie acoustique (152) pour générer un signal acoustique d'élimination,
   une pluralité de canaux de filtre adaptatif (120, 140) réagissant audit signal de détecteur de bruit et audit signal d'erreur, chaque canal étant restreint au fonctionnement sur une sous-bande de fréquence prédéterminée comprenant ladite largeur de bande de bruit et produisant un signal de sortie de canal ; et
   des moyens (150) pour combiner ladite pluralité de signaux de sortie de canal afin de délivrer un signal combiné pour attaquer ledit dispositif de sortie acoustique (152) afin de générer ledit signal acoustique d'élimination,

   caractérisé en ce que :

   chaque canal emploie un retard dans la remise à jour de poids de filtre adaptatif afin d'obtenir une stabilité de fonctionnement dans ladite sous-bande de fréquence sur laquelle fonctionne ledit canal,

   dans lequel les valeurs de retard respectives pour les canaux respectifs sont des valeurs de retard différentes.

2. Système d'élimination selon la revendication 1, **caractérisé** de plus en ce que chaque canal (120, 140) comprend de plus des moyens formant filtre passe-bande (121, 130) pour filtrer ledit signal de détecteur de bruit et ledit signal d'erreur de façon à ne transmettre que les composantes de fréquence de signal à l'intérieur de la sous-bande de fréquence respective pour ledit canal, de façon à restreindre par conséquent ledit canal au fonctionnement sur ladite sous-bande de fréquence.

3. Système d'élimination selon l'une quelconque des revendications précédentes, **caractérisé** de plus en ce que chacun desdits canaux (120, 140) comprend des moyens formant filtre adaptatif récursif (138).

4. Système d'élimination selon l'une quelconque des revendications précédentes, **caractérisé** de plus en ce que lesdites sous-bandes de fréquence couvrent ladite largeur de bande de bruit.

5. Système d'élimination selon l'une quelconque des revendications précédentes, **caractérisé** de plus en ce que chaque canal (120, 140) comprend en outre des moyens de retard (125) pour délivrer une version retardée dudit signal de détecteur de bruit retardée d'un retard prédéterminé, des moyens logiques de remise à jour de poids de filtre adaptatif (126) réagissant à ladite version retardée dudit signal de détecteur de bruit en remettant à jour des entrées de poids de filtre adaptatif vers des moyens formant filtre adaptatif (123) composant ledit canal.

6. Système d'élimination selon la revendication 1, dans lequel ladite pluralité de canaux de filtre adaptatif est de plus **caractérisée par** :

   un premier canal d'élimination (120) couplé audit détecteur de bruit (110) et audit détecteur acoustique (154), ledit premier canal comprenant des premiers moyens formant filtre passe-bande (121) pour filtrer lesdits signaux de détecteur de bruit, ledit premier filtre comportant une première bande passante, des deuxièmes moyens formant filtre passe-bande (130) pour filtrer les signaux générés par ledit détecteur acoustique, ledit deuxième filtre comportant ladite première bande passante, des premiers moyens de retard (125) pour retarder lesdits si-

gnaux de détecteur de bruit filtrés par une première bande passante d'un premier retard de temps présélectionné, et des premiers moyens formant filtre adaptatif comportant une pluralité d'entrées couplées auxdits premier et deuxième moyens formant filtre passe-bande (121, 130) et auxdits premiers moyens de retard (125), et délivrant une première sortie de filtre ; et

un deuxième canal d'élimination couplé audit détecteur de bruit et audit détecteur acoustique, ledit deuxième canal comprenant des troisièmes moyens formant filtre passe-bande pour filtrer lesdits signaux de détecteur de bruit, ledit troisième filtre comportant une deuxième bande passante, des quatrièmes moyens formant filtre passe-bande pour filtrer lesdits signaux de détecteur acoustique, ledit quatrième filtre comportant ladite deuxième bande passante, des deuxièmes moyens de retard pour retarder lesdits signaux de détecteur de bruit filtrés par une troisième bande passante d'un deuxième retard de temps présélectionné, et des deuxièmes moyens formant filtre adaptatif comportant une pluralité d'entrées couplées auxdits deuxièmes moyens formant filtre passe-bande, audit détecteur acoustique et auxdits deuxièmes moyens de retard, et délivrant une deuxième sortie de filtre.

7. Dispositif d'élimination selon la revendication 6, dans lequel lesdits premiers moyens formant filtre adaptatif comprennent une pluralité de premiers poids de filtre, et des premiers moyens logiques de remise à jour de poids (126) réagissant auxdits deuxièmes signaux filtrés par une deuxième bande passante venant dudit détecteur acoustique en ajustant lesdits premiers poids de filtre, lesdits deuxièmes moyens formant filtre adaptatif comprennent une pluralité de deuxièmes poids de filtre, et des deuxièmes moyens logiques de remise à jour de poids réagissant auxdits signaux filtrés par une quatrième bande passante venant dudit détecteur acoustique en ajustant lesdits deuxièmes poids de filtre.

8. Dispositif d'élimination selon la revendication 7, **caractérisé** de plus en ce que ledit premier retard de temps n'est pas égal audit deuxième retard de temps.

9. Système d'élimination selon la revendication 7 ou la revendication 8, **caractérisé** de plus en ce que lesdits premier et deuxième signaux de sortie de filtre sont des signaux numérisés, et en ce que lesdits moyens de combinaison (150) comprennent des moyens formant additionneur numérique.

10. Système d'élimination selon la revendication 7, 8 ou 9, **caractérisé** de plus en ce que lesdits premiers moyens formant filtre adaptatif comprennent des moyens formant filtre adaptatif récursif (138) comprenant :

un premier filtre adaptatif (123) réagissant auxdit signaux de détecteur de bruit filtrés par une première bande passante et comprenant une pluralité de premières entrées de poids de filtre adaptatif, ledit premier filtre adaptatif délivrant une première sortie de filtre adaptatif ;
des premiers moyens logiques de remise à jour de poids (126) réagissant auxdits signaux de détecteur de bruit filtrés par une première bande passante retardés et auxdits signaux de détecteur acoustique filtrés par une deuxième bande passante en remettant à jour de façon adaptative lesdites premières entrées de poids de filtre adaptatif ;
un deuxième filtre adaptatif (132) pour délivrer une deuxième sortie de filtre adaptatif ;
des moyens (124) pour combiner lesdites première et deuxième sorties de filtre adaptatif afin de délivrer ladite première sortie de filtre ;
ledit deuxième filtre adaptatif réagissant à ladite première sortie de filtre et comprenant une pluralité de deuxièmes entrées de poids de filtre adaptatif ;
des troisièmes moyens de retard (133) pour délivrer une version retardée de ladite première sortie de filtre qui est retardée d'un troisième retard de temps prédéterminé ;
des deuxièmes moyens logiques de remise à jour de poids (134) réagissant à ladite version retardée de ladite première sortie de filtre et auxdits signaux de détecteur acoustique filtrés par une deuxième bande passante en remettant à jour de façon adaptative lesdites deuxièmes entrées de poids de filtre adaptatif.

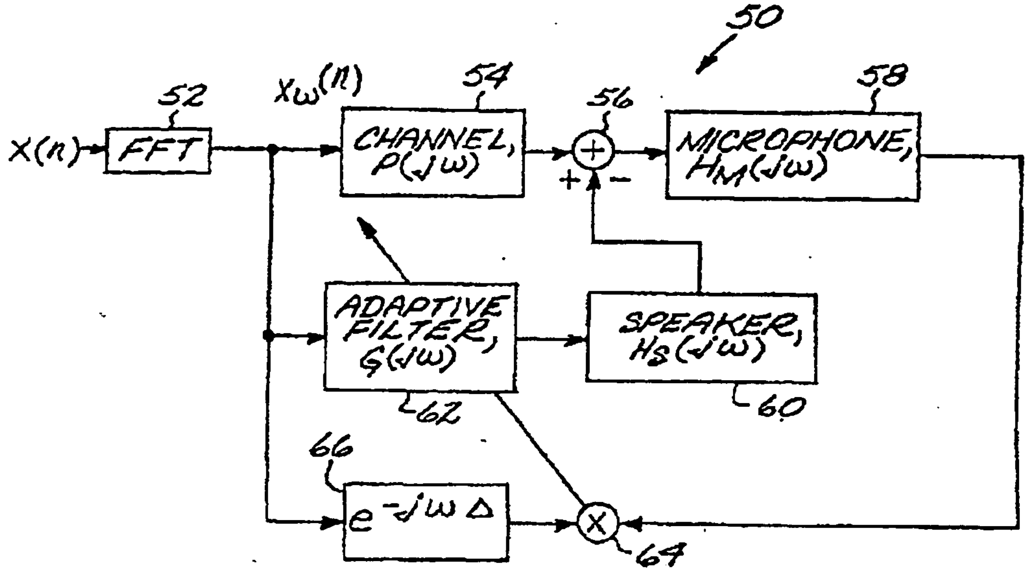
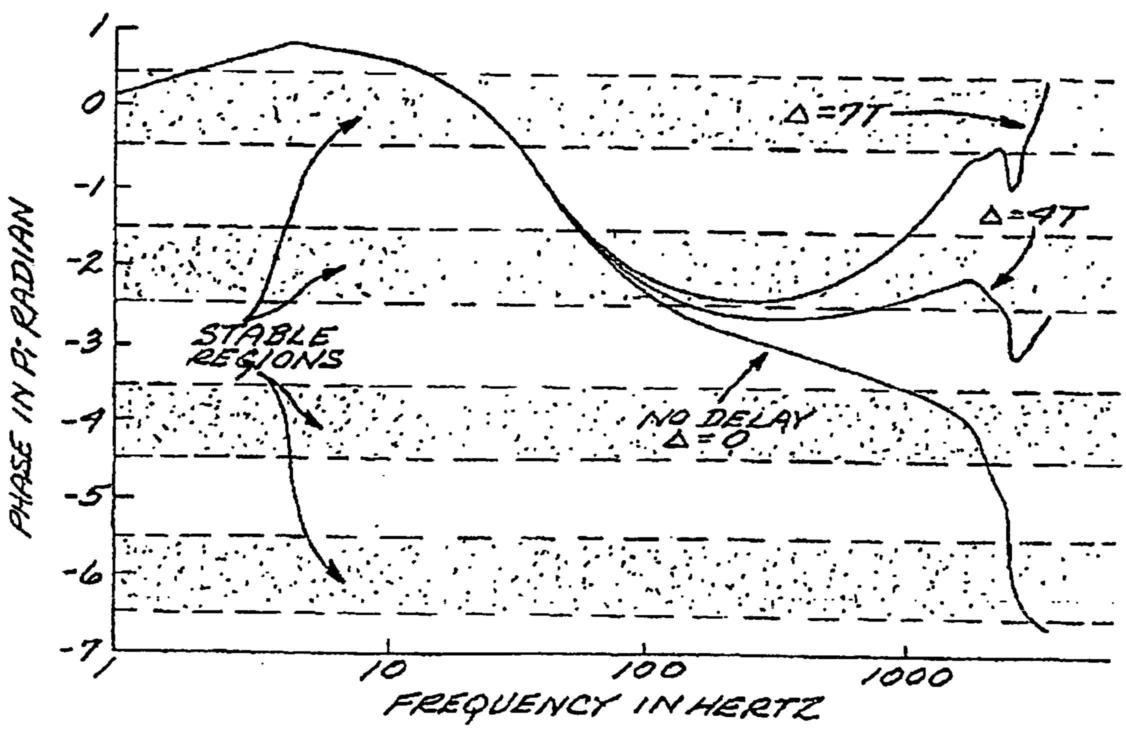

50
52
54
56
58
X(n)
FFT
Xω(n)
CHANNEL, P(jω)
+
-
MICROPHONE, HM(jω)
ADAPTIVE FILTER, G(jω)
SPEAKER, HS(jω)
62
60
66
e-jωΔ
64

FIG. 1

FIG. 2


1
0
-1
-2
-3
-4
-5
-6
-7
PHASE IN PI RADIAN
Δ=7T
Δ=4T
STABLE REGIONS
NO DELAY Δ=0
1
10
100
1000
FREQUENCY IN HERTZ

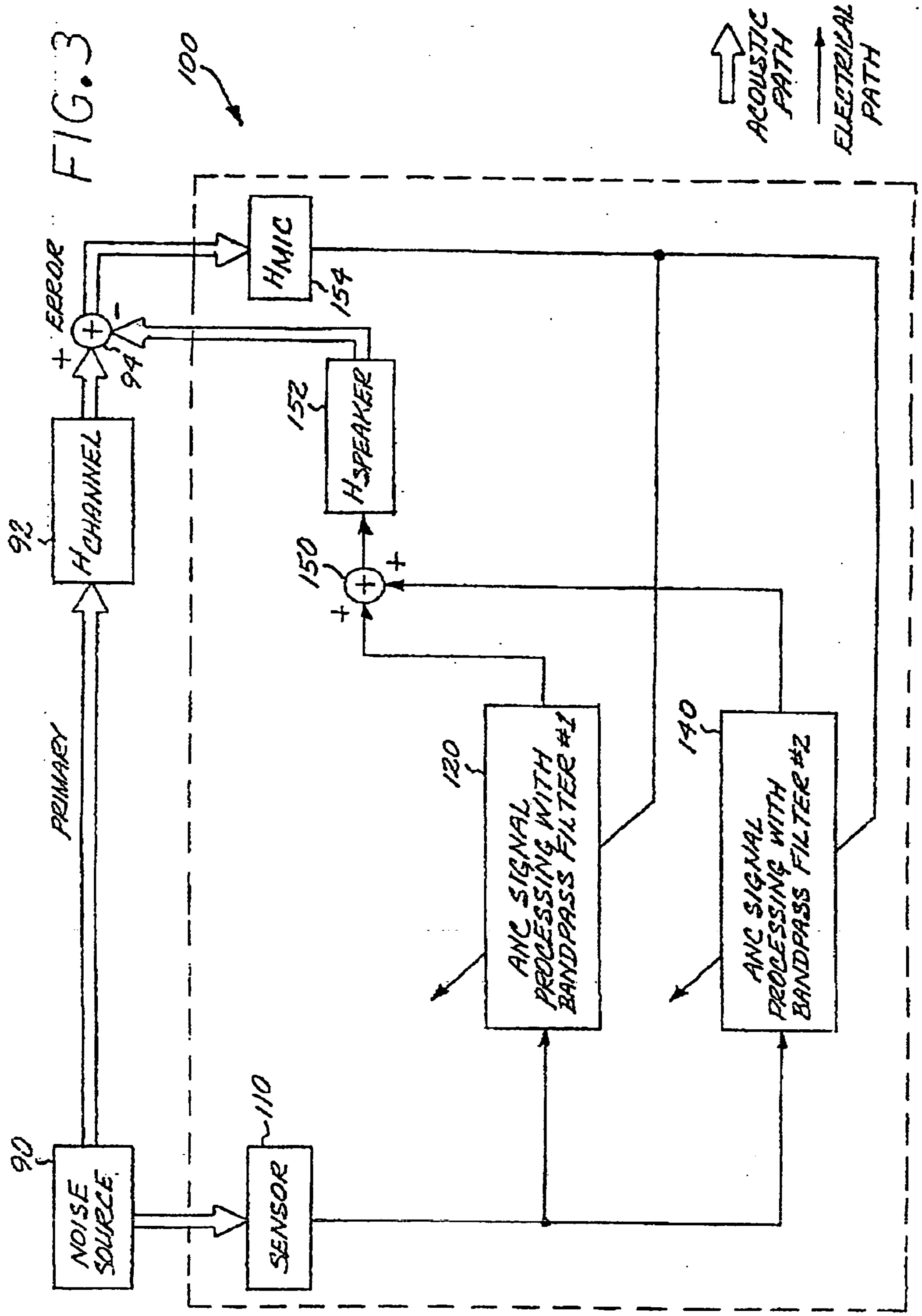


FIG. 3
100
NOISE SOURCE
90
PRIMARY
HCHANNEL
92
+
94
ERROR
-
HMIC
154
SENSOR
110
ANC SIGNAL PROCESSING WITH BANDPASS FILTER #1
120
ANC SIGNAL PROCESSING WITH BANDPASS FILTER #2
140
150
+
+
HSPEAKER
152
ACOUSTIC PATH
ELECTRICAL PATH

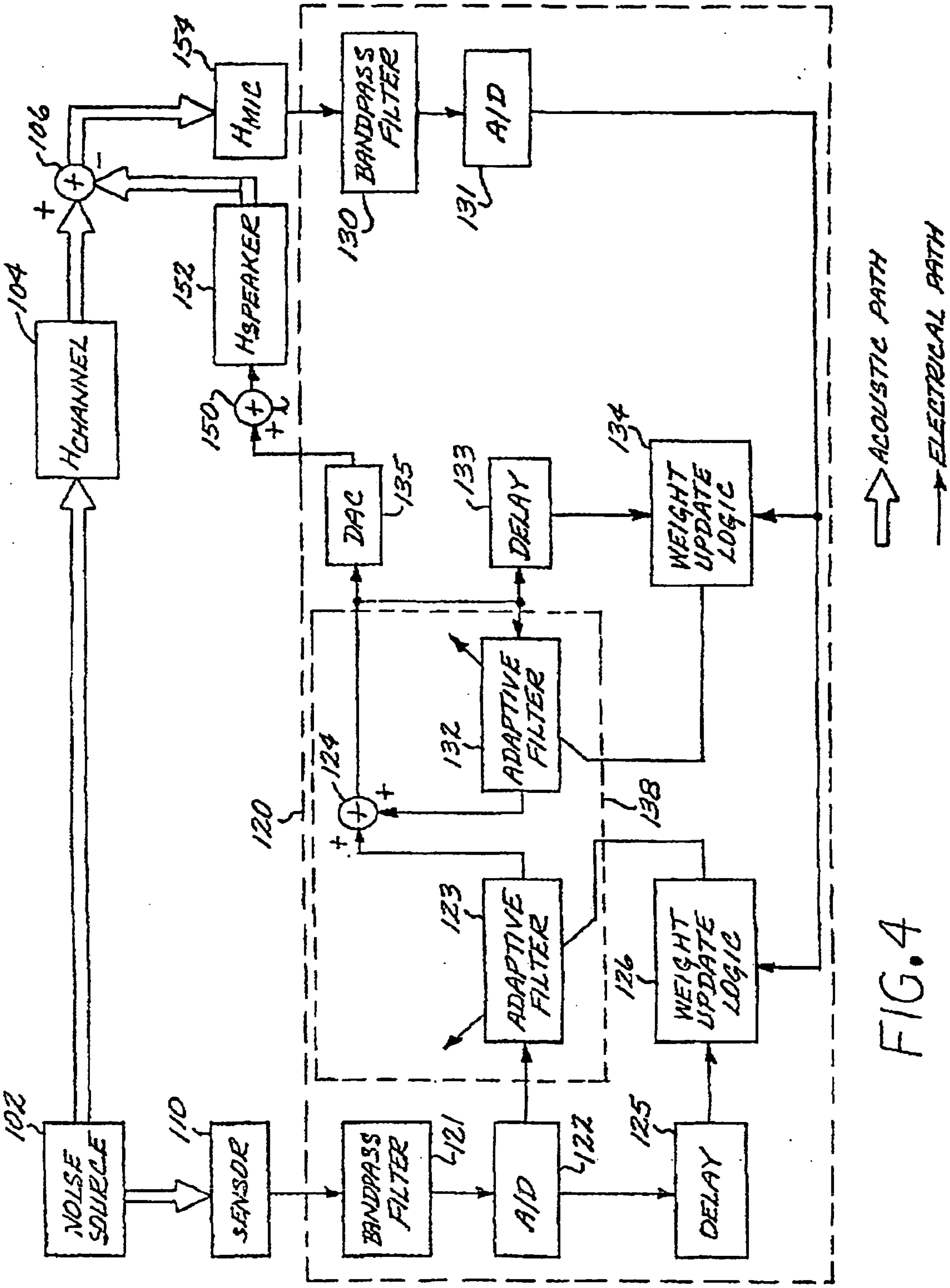

NOISE SOURCE
102
$H_{CHANNEL}$
104
106
H_{MIC}
154
$H_{SPEAKER}$
152
150
SENSOR
110
BANDPASS FILTER
121
A/D
122
DELAY
125
ADAPTIVE FILTER
123
WEIGHT UPDATE LOGIC
126
124
120
ADAPTIVE FILTER
132
138
DAC
135
DELAY
133
WEIGHT UPDATE LOGIC
134
BANDPASS FILTER
130
A/D
131
ACOUSTIC PATH
ELECTRICAL PATH

FIG. 4

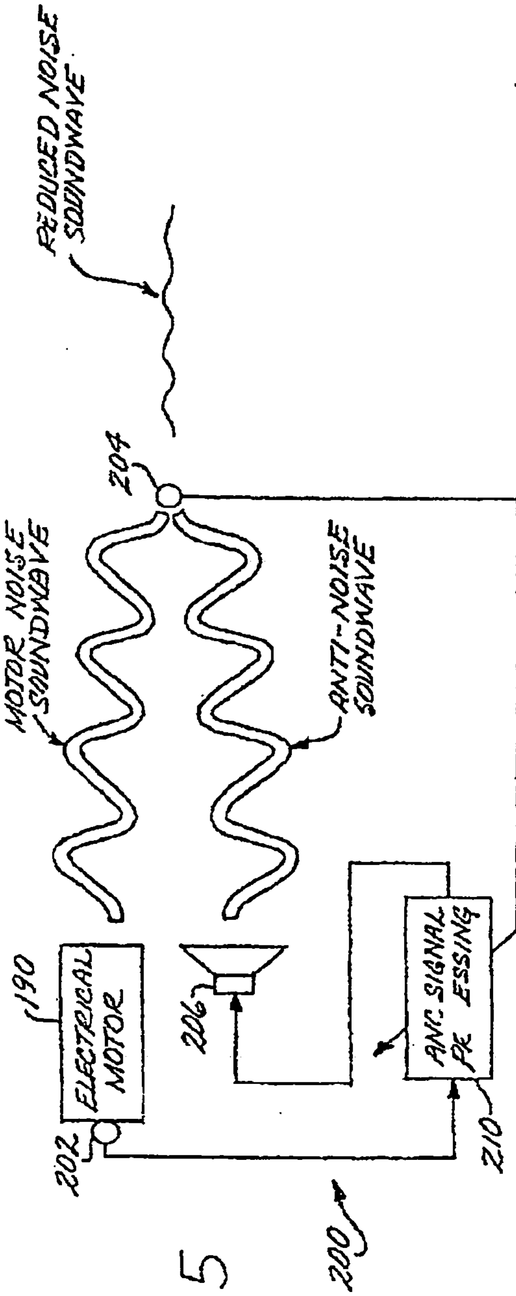

REDUCED NOISE SOUNDWAVE
MOTOR NOISE SOUNDWAVE
ANTI-NOISE SOUNDWAVE
204
190
ELECTRICAL MOTOR
206
ANC SIGNAL PR ESSING
202
210
200

FIG. 5

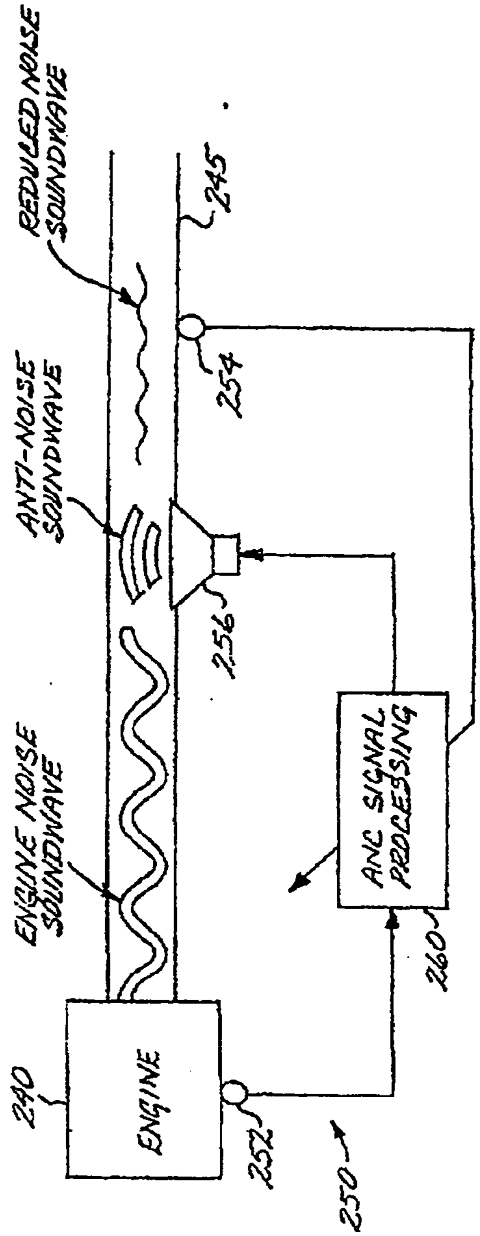

REDUCED NOISE SOUNDWAVE
245
ANTI-NOISE SOUNDWAVE
254
256
ENGINE NOISE SOUNDWAVE
ANC SIGNAL PROCESSING
240
ENGINE
252
260
250

FIG. 6

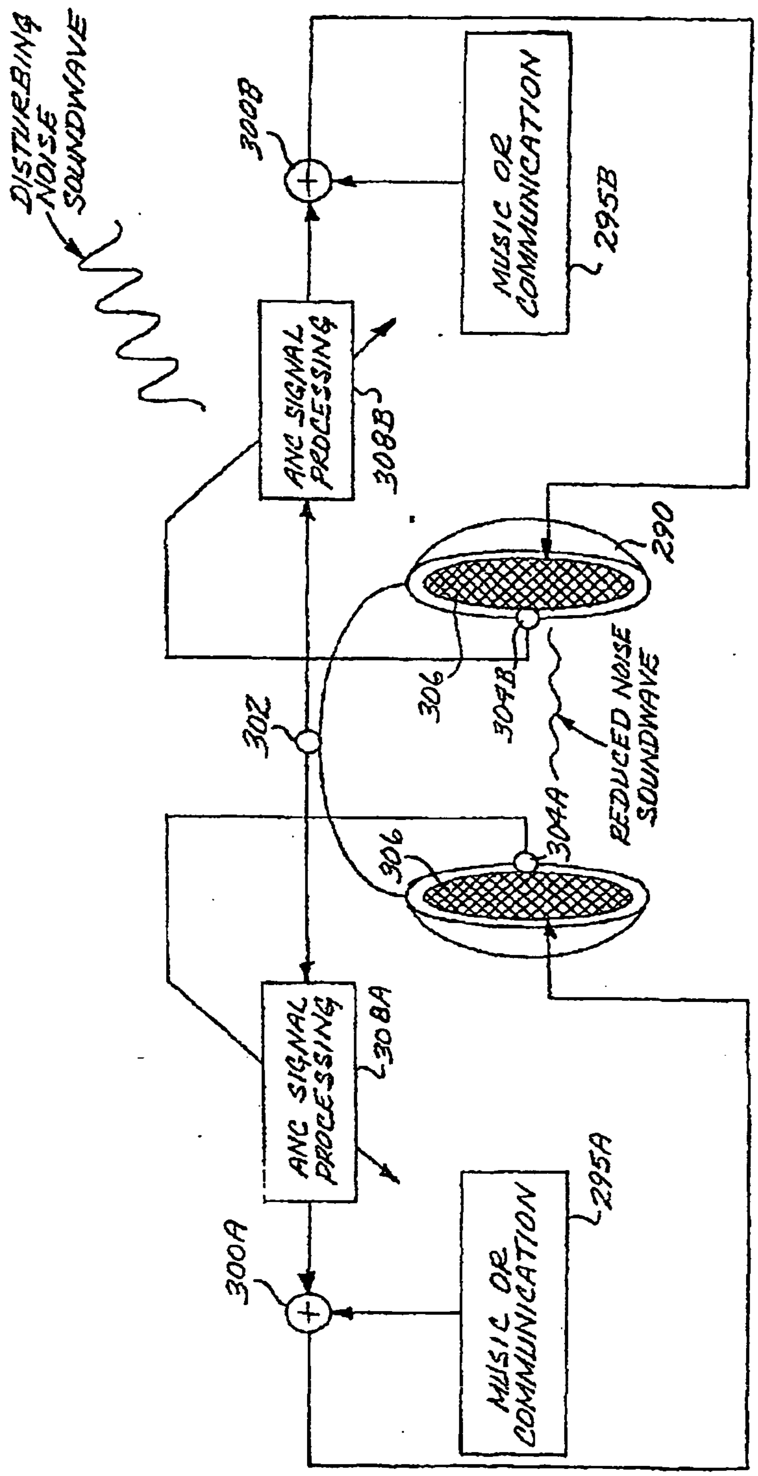

DISTURBING NOISE SOUNDWAVE
300B
ANC SIGNAL PROCESSING
308B
MUSIC OR COMMUNICATION
295B
290
306
304B
302
REDUCED NOISE SOUNDWAVE
304A
306
ANC SIGNAL PROCESSING
308A
MUSIC OR COMMUNICATION
295A
300A

FIG. 7